Listing of Claims

This listing of claims will replace all prior versions and listings of claims in the application.

Claim 1 (previously presented): A method of processing sound data, wherein, before a playback of the sound by a playback device:

a) signals representative of at least one sound propagating in a three-dimensional space and arising from a source situated at a first distance from a reference point are coded so as to obtain a representation of the sound by components expressed in a base of spherical harmonics, of origin corresponding to said reference point,

b) and a compensation of a near field effect is applied to said components by a filtering which is dependent on a second distance defining substantially , for a playback of the sound by said playback device, a distance between a playback point and a point of auditory perception.

Claim 2 (previously presented): The method as claimed in claim 1, wherein, said source being far removed from the reference point:

- components of successive orders m are obtained for the representation of the sound in said base of spherical harmonics, and
- a filter is applied, the coefficients of which, each applied to a component of order m, are expressed analytically in the form of the inverse of a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance, so as to compensate for a near field effect at the level of the playback device.

Claim 3 (previously presented): The method as claimed in claim 1, wherein, said source being a virtual source envisaged at said first distance:

- components of successive orders m are obtained for the representation of the sound in said base of spherical harmonics, and
- a global filter is applied, the coefficients of which, each applied to a component of order m, are expressed analytically in the form of a fraction, in which:
    - the numerator is a polynomial of power m, whose variable is inversely

proportional to the sound frequency and to said first distance, so as to simulate a near field effect of the virtual source, and

- the denominator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance, so as to compensate for the effect of the near field of the virtual source in the low sound frequencies.

Claim 4 (previously presented): The method as claimed in claim 1, wherein the data coded and filtered in steps a) and b) are transmitted to the playback device with a parameter representative of said second distance.

Claim 5 (previously presented): The method as claimed in claim 1 wherein, the data coded and filtered in steps a) and b) are stored with a parameter representative of said second distance on a memory medium intended to be read by the playback device.

Claim 6 (previously presented): The method as claimed in claim 4, in which, prior to a sound playback by a playback device comprising a plurality of loudspeakers disposed at a third distance from said point of auditory perception, an adaptation filter whose coefficients are dependent on said second and third distances is applied to the coded and filtered data.

Claim 7 (previously presented): The method as claimed in claim 6, wherein the coefficients of said adaptation filter, each applied to a component of order m, are expressed analytically in the form of a fraction, in which:

- the numerator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance,
- and the denominator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said third distance.

Claim 8 (previously presented): The method as claimed in claim 2, wherein, for the implementation of step b), there is provided:

- in respect of the components of even order m, audiodigital filters in the form of a cascade of cells of order two; and

- in respect of the components of odd order m, audiodigital filters in the form of a cascade of cells of order two and an additional cell of order one.

Claim 9 (previously presented): The method as claimed in claim 8, wherein the coefficients of an audiodigital filter, for a component of order m, are defined from the numerical values of the roots of said polynomials of power m.

Claim 10 (previously presented): The method as claimed in claim 2, wherein said polynomials are Bessel polynomials.

Claim 11 (previously presented): The method as claimed in claim 1, wherein there is provided a microphone comprising an array of acoustic transducers arranged substantially on the surface of a sphere whose center corresponds substantially to said reference point, so as to obtain said signals representative of at least one sound propagating in the three-dimensional space.

Claim 12 (previously presented): The method as claimed in claim 11, wherein a global filter is applied in step b) so as, on the one hand, to compensate for a near field effect as a function of said second distance and, on the other hand, to equalize the signals arising from the transducers so as to compensate for a weighting of directivity of said transducers.

Claim 13 (previously presented): The method as claimed in claim 11 wherein there is provided a number of transducers that depends on a total number of components chosen to represent the sound in said base of spherical harmonics.

Claim 14 (previously presented): The method as claimed in claim 1, in which in step a) a total number of components is chosen from the base of spherical harmonics so as to obtain, on playback, a region of the space around the point of perception in which the playback of the sound is faithful and whose dimensions are increasing with the total number of components.

Claim 15 (previously presented): The method as claimed in claim 14, wherein there is

provided a playback device comprising a number of loudspeakers at least equal to said total number of components.

Claim 16 (previously presented): The method as claimed in claim 1, wherein:

- there is provided a playback device comprising at least a first and a second loudspeaker disposed at a chosen distance from a listener,
- a cue of awareness of the position in space of sound sources situated at a predetermined reference distance from the listener is obtained for this listener, and
- the compensation of step b) is applied with said reference distance substantially as second distance.

Claim 17 (previously presented): The method as claimed in claim 4, wherein:

- there is provided a playback device comprising at least a first and a second loudspeaker disposed at a chosen distance from a listener,
- a cue of awareness of the position in space of sound sources situated at a predetermined reference distance from the listener is obtained for this listener, and
- prior to a sound playback by the playback device, an adaptation filter whose coefficients are dependent on the second distance and substantially on the reference distance, is applied to the data coded and filtered in steps a) and b).

Claim 18 (previously presented): The method as claimed in claim 16, wherein:

- the playback device comprises a headset with two headphones for the respective ears of the listener, and
- separately for each headphone, the coding and the filtering of steps a) and b) are applied with regard to respective signals intended to be fed to each headphone, with, as first distance, respectively a distance separating each ear from a position of a source to be played back.

Claim 19 (previously presented): The method as claimed in claim 1, wherein a matrix system is fashioned, in steps a) and b), said system comprising at least:

- a matrix comprising said components in the base of spherical harmonics, and

- a diagonal matrix whose coefficients correspond to filtering coefficients of step b),

and said matrices are multiplied to obtain a result matrix of compensated components.

Claim 20 (previously presented): The method as claimed in claim 19, wherein:

- the playback device comprises a plurality of loudspeakers disposed substantially at one and the same distance from the point of auditory perception, and
- to decode said data coded and filtered in steps a) and b) and to form signals suitable for feeding said loudspeakers:

  * a matrix system is formed comprising said result matrix and a predetermined decoding matrix, specific to the playback device, and

  * a matrix is obtained comprising coefficients representative of the loudspeakers feed signals by multiplication of the matrix of the compensated components by said decoding matrix.

Claim 21 (previously presented): A sound acquisition device, comprising a microphone furnished with an array of acoustic transducers disposed substantially on the surface of a sphere, wherein the device furthermore comprises a processing unit arranged so as to:

- receive signals each emanating from a transducer,
- apply a coding to said signals so as to obtain a representation of the sound by components expressed in a base of spherical harmonics, of origin corresponding to the center of said sphere,
- and apply a filtering to said components, which filtering is dependent, on the one hand, on a distance corresponding to the radius of the sphere and, on the other hand, on a reference distance.

Claim 22 (previously presented): The device as claimed in claim 21, wherein said filtering consists, on the one hand, in equalizing, as a function of the radius of the sphere, the signals arising from the transducers so as to compensate for a weighting of directivity of said transducers and, on the other hand, in compensating for a near field effect as a function of a chosen reference distance, defining substantially, for a playback of the sound, a distance between a playback point and a point of auditory perception.